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Simultaneous subliminal signalling in conventional sound circuits:
a feasibility study

No. 1971/1

Research Department, Engineering Division
THE BRITISH BROADCASTING CORPORATION

RESEARCH DEPARTMENT

SIMULTANEOUS SUBLIMINAL SIGNALLING IN CONVENTIONAL SOUND CIRCUITS: A FEASIBILITY STUDY

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SIMULTANEOUS SUBLIMINAL SIGNALLING IN CONVENTIONAL SOUND CIRCUITS: A FEASIBILITY STUDY

Summary

Programme transmission networks use auxilliary lines for operational control. Considerable savings might be possible if the auxilliary signals could be combined with the associated sound signal and sent simultaneously on a single circuit. The simultaneous system should operate in-band and provide sufficient information for operational use. Ideally, it should be possible for the combined signal to be directly broadcast with negligible programme impairment and without further processing at the receiving terminal.

Some fundamental methods for simultaneous subliminal signalling have been considered theoretically and the most promising systems investigated experimentally. The results show that signalling systems based on 'frequency-notches' are worth further investigation.

1. Introduction

The transmission of programme signals through the sound networks requires continual monitoring and technical control; in the case of temporary contribution networks, there is also the requirement of providing performance testing and talk-back facilities. These facilities demand the use of auxilliary circuits to carry the extra programme-related information; normal practice is to employ additional sound lines and telephone links for this purpose. Worthwhile economies might be made if some or all of these channels could be provided in the main sound circuit by suitably combining the extra information with the programme signal itself; this process will be referred to as simultaneous signalling.

The programme signal normally uses the total bandwidth provided by the sound circuit, so that any simultaneous signalling process must be essentially 'in-band'; in general the bandwidth of sound circuits cannot be extended. The signalling information should be available at any point in the sound network. It would also be an advantage if the combined signal could be broadcast without removing the additional signals as this would enable unattended transmitters fed by RBL to be controlled and maintained. The last condition can only be met by making the added signals totally 'subliminal'; that is, the presence of the additional signals should not produce any noticeable impairment.

The requirement is therefore for a simultaneous in-band signalling system which is subliminal to the most discriminating listener, whilst providing enough signalling capacity to meet operational needs. The fundamental aspects of several signalling methods were considered theoretically and those systems which looked promising were then investigated experimentally. The object of the experimental work was not to design an actual system, but to estimate the maximum subliminal signalling rates of the methods investigated. The present report surveys these methods and describes the results of the experimental tests.

2. Survey of possible signalling methods

2.1. General considerations

There are various limitations and imperfections in human hearing which it might be possible to exploit in devising a subliminal signalling system. Any such system will be based on some form of redundancy in order to incorporate additional information within the available channel. Exploiting the limitations in human hearing, however, is not expected to lead to signalling systems with substantial rates of information because the ear is a remarkably acute instrument; by contrast the limitations in human vision which have made present-day television systems possible are vastly greater.

Some of the more important aural limitations* which were considered for the present application are given below; the list is not claimed to be exhaustive and the order of presentation is arbitrary.

a) Frequency weighting characteristics of human hearing (Fletcher-Munsen audibility curves[1]); the subjective loudness of individual sound components depends jointly upon their intensity and frequency. For example, at very low sound intensities near to the audibility threshold, the ear's sensitivity at 50 Hz is about 50 dB less than that at the most sensitive frequency (about 2·5 kHz).

b) Insensitivity of the ear to phase information[2] (Ohms law of hearing); This is normally understood to imply that the ear can perceive amplitude and frequency but not phase. This formulation must break down if phase-changes significantly modify the actual envelope of the signal. For complex sound signals, dispersion should probably not be allowed to exceed about 8 msecs between any two frequency components; this figure is the presently quoted upper limit in group-delay difference between the maximum usable frequency and the band-centre for high-quality music lines.[3]

* monophonic sound signals are assumed here.

c) Sound Masking Phenomena;[4] in its simplest form, this effect shows itself as the suppression of quiet sounds by relatively loud sounds of comparable frequency. The degree of masking is generally reduced if the frequency separation between quiet and loud components increases.

d) Subjective Tone Generation;[5] the non-linearity of the ear can generate extra subjective sound components which are not present in the input waveform. If the input signal comprises simple tones, the extra components are harmonic tones and also sum and difference frequencies.*

e) Theory of Missing Fundamentals;[6] this effect is a type of dual of (d) above. If the fundamental frequency component is omitted from a complex sound, the presence of the related harmonics often allows it to be heard subjectively.

f) Time Discrimination (Haas effect);[7] this relates to the 'echo tolerance' of the ear and, although under special experimental conditions, time intervals of about 10 msec can be noticed, it is normally only possible to discriminate the echo if the interval is more than about 25 msec.

g) Frequency Response Tolerance; the ear can apparently tolerate small perturbations in the amplitude-frequency response of the sound channel but there is little published information on the subject. Intelligibility measurements on speech[8] have revealed that spectral humps reduce intelligibility more than corresponding spectral depressions. Results obtained from work on loudspeakers have shown that variations of about ±2 dB over the audio band can be tolerated; the ear is more sensitive than this, however, in respect of smooth amplitude slopes across the spectrum.

h) Programme Drop-out Tolerance; investigations into the audibility of tape drop-outs in magnetically recorded sound signals[9] have shown that regular low frequency (<2 Hz) attenuation notches of reasonable depth (>20 dB) can be inaudible if their duration time is less than about 1–2 msecs. The ear is rather more tolerant than this to isolated random drop-outs.

i) Frequency Tolerance; this is a very small effect because the ear is astonishingly sensitive to frequency changes. The minimum perceptible frequency change varies with sound level and is between 2 and 4 Hz over the lower part of the audio spectrum but beyond about 2 kHz[10] the minimum perceptible shift is an approximately constant fraction of the nominal frequency. (A feature of f.d.m. carrier circuits is that the entire sound spectrum can be subject to small frequency shifts; recent subjective work on this topic[11] has shown that a maximum shift of ±2 Hz is normally acceptable.)

j) Intelligibility of Frequency-compressed Speech; work on the analysis and synthesis of speech signals in connection with Vocoders[12] has revealed some interesting possibilities which may be applicable to the present problem. The fundamental principle behind the operation of most Vocoder channels is the assumption that the ear behaves as a short-term frequency analyser so that the information may be transmitted as packets of 'elementary' time-frequency signals.[13] The characteristics of speech are such that, for acceptable intelligibility, only a limited number of these elementary signals need to be sent; moreover, speech signals may be further compressed by coarse quantisation and, in the limit, infinitely clipped (i.e. unity-bit) signals can be sent without complete loss of intelligibility.[14] These results, however, do not necessarily apply to the other types of programme material.

Broadly speaking, it should be possible to exploit each of the audio effects, (a) to (j) above for simultaneous signalling in one of two fundamental ways. The extra signalling components may be added to the sound signal, – an 'additive' system – or, the sound signal itself may be modulated in some way to incorporate the extra components – a 'multiplicative' system. It is also possible to envisage 'hybrid' systems which utilise both additive and multiplicative components. Multiplicative systems are expected to offer somewhat higher subliminal signalling speeds than additive systems but will be generally more difficult to instrument. They are also programme-dependent and therefore of variable signalling speed – multiplicative signalling cannot take place in the absence of programme. The signalling information will normally need to be coded before it can be combined with the sound signal; the information can then be extracted at any point on the transmission link by detection and decoding.

The general arrangement of a simultaneous signalling system is given in Fig. 1. Insertion of the signalling information, in coded form, into the sound programme signal by the combining unit is assumed to take place at baseband frequencies; the coding system for the signalling information is most likely to be digital. The synchronisation facility shown in Fig. 1 will be required if the information rate has to be programme-dependent for subliminal reasons or if 'word' synchronisation is necessary. For additive systems the combining unit is simply an adder but for multiplicative systems a more complicated signal processing unit (e.g. a modulator) is required.

If the system is not fully subliminal, the signalling information may have to be suppressed at the receiving end of the link or, with some types of signalling, the sound signal might have to be corrected or 'repaired'. The process is indicated by the short broken lines in Fig. 1.

Many theoretical subliminal signalling methods were considered. Table 1 summarises these and the following Sections in the Report describe them in rather greater detail; some of the proposed signalling methods are believed to be novel. It is worth pointing out that whilst those systems which produce programme-like interference should give least impairment, they will undoubtedly be the most difficult to detect in presence of programme.

2.2. Additive systems

The simplest signalling method of this type employs one or more suitably modulated low-level tones (systems 1 and 2, Table 1); the presence or absence of a single tone

* these components are sometimes referred to as audible 'beats'.

can carry one 'bit' of information. The highest subliminal level for modulated tones occurs at the extremities of the audio spectrum, where the ear's sensitivity is minimal. In the past, such methods have been considered to be impractical because the band-edge performance of sound lines in respect of amplitude and phase equalisation is normally very ill-defined and the increased possibility of overload with additional signals cannot be ignored. However a signalling system working at the lower end of the spectrum for remotely switching stereo-coders at transmitters has already been reported.[15]

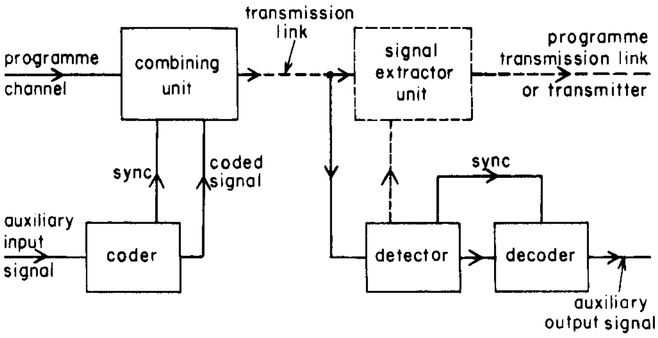

Fig. 1 - Generalised system for simultaneous subliminal signalling in sound circuits

In the presence of programme, suitably chosen tone and pulse signals can be masked[16] to some extent by the actual programme material (system 3). Moreover, the signals could be matched to the 'running spectrum' of the programme, thus increasing the level at which they may be subliminally introduced.

A slow but potentially reliable signalling system can be derived by superimposing long low-level pseudo-random binary noise sequences[17] which will produce the same audible effect as ordinary Gaussian noise and should therefore be subjectively acceptable (system 4). (The subjective impairment produced by different types of audio noise has been assessed recently and the results are given in Reference 18.) A possible practical embodiment inserts a maximal length sequence (m-sequence) for signalling a binary 'one' and omits, or changes, the sequence for a binary 'zero'; the detection process involves correlating the received signal with a locally-generated m-sequence.

Also included in Table 1 is a method (system 5) in which band-limited Gaussian noise is actually added to the programme and signalling is achieved by frequency-weighting the added noise.

TABLE 1

Some Theoretical Methods for Simultaneous Subliminal Signalling in Sound Circuits

SYSTEM TYPE AND NUMBER		SIGNALLING SYSTEM	MODULATION††	SUBLIMINAL BASIS*
ADDITIVE	1.	Low level l.f. tone	pulse *or* f.s.k.	(a)
	2.	Low level h.f. tone	pulse *or* f.s.k.	(a) (d)
	3.	Masked signals	analogue *or* digital	(c)
	4.	Noise signals	pseudo-random binary	(c)
	5.	Spectrum perturbation of added noise	binary switching	(c) (g)
MULTIPLICATIVE	6.	Signal-phase switching	phase modulation	(b)
	7.	Constant envelope	phase *or* binary p.c.m.	(j) (b)
	8.	Amplitude drop-outs†	p.c.m.	(h)
	9.	Frequency notches†	multilevel p.c.m.	(e) (g)
	10.	Quantisation switching†	binary p.c.m.	(j)
	11.	Programme frequency perturbation	f.s.k.	(b) (i)
	12.	Artificial Reverberation†	binary p.c.m.	(f)
HYBRID	13.	Tone-burst synchronisation	binary	(a) (d)
	14.	Amplitude drop-out insert†	binary p.c.m.	(h) (j)
	15.	Frequency-notch insert†	multilevel p.c.m.	(e) (g) (a)
	16.	Quadrature phase signals	a.m., p.m., *or* p.c.m.	(a) (b)

* classification refers to the list of aural limitations given in Section 2.1.

† these systems were selected for experimental assessment (Section 3).

†† f.s.k. is frequency-shift keying; p.c.m. is pulse-code modulation; a.m. is amplitude modulation
p.m. is phase modulation

2.3. Multiplicative systems

Multiplicative signalling systems employ modulation of the sound signal itself and are necessarily programme-dependent; in some cases it may be advantageous to restrict the modulation to selected bands of frequencies in the sound channel. The great drawback with multiplicative signalling is that sound signals are unpredictable and modulated and unmodulated parts are quite likely to be confused.

Perhaps the most tantalising characteristic of sound waveforms as far as the present application is concerned is their apparent phase redundancy; it should be possible, theoretically at least, to evolve a signalling system based on this particular limitation in hearing. Phase-switching methods (system 6, Table 1) can be envisaged which retain the original power spectrum of the sound waveform but modify its phase characteristic in a way detectable by suitable circuits.[19] Similar communication systems employing orthogonal time functions have also been proposed.[13,20]

There are however, many practical drawbacks. First, sound waveforms are quite likely to contain inherent phase patterns somewhat similar to those required for signalling; second, the phase equalisation of most audible circuits would not be sufficiently well controlled for reliable signalling. Third, and more fundamental, the process of phase modulation, unless performed very slowly, would itself introduce unwanted signal components. Finally because unlimited phase dispersion is not subjectively tolerable (Section 2.1(b)), the process would probably have to be restricted to the upper end of the audio spectrum and this could severely limit the signalling rate with predominantly low-frequency programme material.

In a further method of signalling, somewhat similar in principle to the phase-switching system described above, high-frequency components of the programme waveform are used as 'carriers' and are phase or frequency-modulated whilst still retaining their original envelope waveshapes; this system will be referred to as 'constant envelope' signalling (system 7).

A method of signalling which appears to be rather more practical is based on the low audibility of short-duration amplitude reduction[9,21] (system 8). Using this principle, binary-coded signalling can be effected by the presence or absence of short periods of attenuation of the programme often referred to as 'drop-outs'. The main problem is to detect drop-outs in sound signals which themselves contain similar phenomena. It might be feasible to insert pulses into the programme drop-outs both to facilitate the detection process and to reduce the subjective impairments. The application of error-detecting (and correcting) codes[22] to combat programme interference might prove necessary.

Another method uses frequency-notches (or by analogy with the previous method – spectral 'drop-outs'). In one form of 'frequency-notch' signalling (system 9) a binary word would be represented by a comb of narrow notches placed in the power spectrum of the sound signal; it is assumed that such combs can be made subliminal and yet also be detected. A serious limitation of this method is that a typical short period of programme material would not cover the spectrum sufficiently well to provide acceptable signalling.

Other multiplicative systems can also be derived; for example, one method involves coded low-level harmonic distortion of the sound waveform. In a second method of this kind, the sound signal is periodically either quantised for a short duration, or left in original analogue form, in order to signify binary-coded information (system 10). Quantisation would have to be relatively coarse (i.e. employ a small number of levels) in order to make it detectable since the modulated sound signal would be subject to the usual bandwidth limitations in the transmission links.

The tolerance of the ear to frequency-shifts, although very small could lead to some interesting theoretical signalling possibilities. The basis of these methods is to shift digitally the frequencies of components in the sound waveform (frequency-shift keying) and then to detect the auxiliary signal by frequency analysis. A simple embodiment of this idea is to perturb the exact mathematical relationship between the most prominant fundamental component in the sound waveform and its harmonic frequencies (system 11). The practical difficulties inherent in this method are very severe however; for example, both sending and receiving terminals require tracking frequency-analysers of extremely high resolution (perhaps of the order of 0·1% bandwidth).

The final multiplicative system of signalling to be described depends on the echo-tolerance of the ear; this is the method of artificial reverberation (system 12) and could provide a fairly high rate of signalling. The process entails adding to the original sound signal a weak echo delayed by say, 2 to 10 ms; the reverberation pattern would represent the data stream of the auxiliary signal. At the receiving terminal the detection process could use cross-correlation methods (i.e. time-shift and multiply) but alternatively it might be possible to use more sophisticated techniques; for example, homomorphic filtering in which the echoes are detected by generating the signal 'cepstrum'.[24] To facilitate detection it might be necessary to employ more complex echo signals such as, for example, doublets, multiple echoes, or even echo-position modulation.

2.4. Hybrid systems

With additive systems, the subliminal constraint would make difficult the error-free detection of signals in the presence of programme; unless the signalling amplitude were made proportional to programme level, the signal level would be determined by its audibility in the very quietest passages. Multiplicative systems on the other hand suffer the disadvantage of offering signalling rates which are programme dependent; moreover, they require rather more sophisticated instrumentation for their realisation. 'Hybrid' systems reduce these difficulties by using both types of signalling simultaneously. In these systems the multiplicative component is made to suppress the programme signal

during periods of additive low-level signalling; for subliminal operation, the interference from both components must be below threshold. A hybrid arrangement can also be used to provide synchronisation for a multiplicative signalling system. Some examples of hybrid signalling systems are listed in Table 1 (13 to 16).

Amplitude drop-outs may also be used to provide the necessary time-slots for pulse or tone-burst signals; the absence of programme during these periods would facilitate the detection process (system 14).

A somewhat similar hybrid signalling system can be evolved using frequency notches. A frequency notch, or comb of such notches, can be permanently 'carved' into the spectrum of the sound signal and a low-level modulated tone signal (which represents one binary digit) inserted at each notch frequency. This is here described as multitone frequency-notch signalling (system 15). The bandwidth of each notch must be sufficiently large to pass the auxiliary modulation and the number and width of such notches which can be subjectively tolerated clearly sets the available information rate.

Another hybrid method, based on the phase-switching principle (system 6), is to convert the high-frequency components of the sound signal into an entirely even (or odd) symmetric waveform without changing its power spectrum and then to use the resulting, empty, high-frequency quadrature channel* to carry the extra signalling information at subliminal level (system 16).

3. Experimental assessment

3.1. General

The object of this part of the work was not to design an actual signalling system but rather to assess experimentally various signalling methods in terms of information capacity and programme impairment. Of the methods discussed in Section 2, the following were thought to warrant experimental investigation:

a) Quantisation signalling (System 10)
b) Reverberation signalling (System 12)
c) Attenuation signalling (Systems 8 and 14)
d) Frequency-notch signalling (Systems 9 and 15)

Most of the time was devoted to examining systems (c) and (d) as these gave more promising results after initial tests; systems (a) and (b) were not investigated in depth. The experimental work was restricted to assessing the subjective impairments introduced by each signalling system; a range of parameters was explored in each case so that the optimum conditions for subliminal signalling could be established.

Unless stated otherwise, the subjective tests took place under the following conditions. A high-quality monitoring loudspeaker was used at normal listening level in a laboratory which, although not acoustically treated, had been arranged to minimise the level of extraneous sounds. The observers were drawn from a group of twelve fairly experienced technical staff; the scoring was based upon the EBU Impairment and Comparative Scales (Tables 2 and 3). The test routine was normally of the ABA type in which the test passage (B) was sandwiched between two unprocessed verions (A). In all tests, the observers were also made to score nominally unimpaired (i.e. unprocessed) programme; this reference score is recorded in the following test results (Figs. 3 and 6 to 11) as 'unprocessed scores'.

The tests were limited to tape-recorded monophonic signals and the programme material included piano music (Schubert Sonata in B flat major, No. D.960), male speech, and popular dance music. The recordings and replay recorder were of professional quality and nominally capable of a bandwidth of 15 kHz. Due to the different instrumentation required for each system, the signal-to-noise ratio of the reproduced programme was a somewhat variable quantity but this figure is quoted where relevant to the results. The subjective effects of signal pre- and de-emphasis and also companding on the signalling systems was not investigated here. Unless stated otherwise, the following results refer to high-quality 15 kHz sound signals.

TABLE 2

E.B.U. Impairment Grades

1. Imperceptible
2. Just perceptible
3. Definitely perceptible but not disturbing
4. Somewhat objectionable
5. Definitely objectionable
6. Unusable

TABLE 3

*E.B.U. Comparative Grades**

+3	Much better than
+2	Better than
+1	Slightly better than
0	Same as
−1	Slightly worse than
−2	Worse than
−3	Much worse than

3.2. Quantisation signalling

The object of the experimental work was to explore the subjective effects of inserting short regular bursts of coarse quantisation into sound signals.

* Time reversal techniques[25] might be used to remove the phase information from the sound signal.

* This scoring scale is given in 'Report of the EBU Ad-Hoc Group on Colour Television', 2nd edition, Feb. 1965.

It was convenient to quantise the audio signal by using an analogue-to-digital converter (a.d.c.) capable of coding a large number of levels and a complementary digital-to-analogue converter (d.a.c.). The experimental arrangements provided adequate resolution up to 10 digits (1024 levels); the unweighted peak signal-to-r.m.s. noise of the output analogue signal (due to quantising noise) was about 65 dB and the subjective impairment of the system was slight. A gate circuit normally passed all 10 digits to the d.a.c., but while it received a signalling pulse, only the first few most significant digits were allowed to reach the d.a.c. In the tests, the number of digits contributing to the coarsely quantised words was progressively reduced until, in the limit, two-level (i.e. unity bit) signals were obtained; the duration and repetition rate of the signalling pulse were also varied.

From initial tests, it soon became clear that very coarse quantisation was unacceptable and a relatively large number of levels would have to be used. For acceptably low impairment with most programme material, the number of levels had to be greater than eight, the burst of quantisation not longer than 4 ms, and the maximum repetition rate 0·5 Hz. When the number of quantum levels was reduced to a practical figure for signalling (say 6) the interference to programme became totally unacceptable for all practical quantising durations and repetition rates.

The results were compared briefly with those obtained by switching into the analogue (actually quantised with 1024 levels) sound waveform comparable amounts of white Gaussian noise. It was generally found that, although quantising distortion produces a sound rather similar to random noise, its effect was no more acceptable than noise of the same r.m.s. value.

The results showed that the ear cannot tolerate short-duration, coarse quantising, so that signalling by switched quantising appears to be unworkable.

3.3. Reverberation signalling

The object here was to determine the sensitivity of the ear to echo signals switched into and out of programme signals. The experimental arrangement used for the tests is shown in Fig. 2. This circuit provided electronically switched signal echoes; the adjustable delay units were phase-equalised to only 10 kHz, but this limitation was thought to be unimportant. The main tests explored the effects of single echoes and antiphase (zero-sum) doublets.

In order to limit the number of subjective tests to a sensible figure the following parameters were fixed:

a) a regular echo-switching pattern was used; this had been found to represent the worse case.*

b) The square-wave switching rate was set to 25 Hz which corresponds to good teleprinter signalling speed (50 bauds).

c) The rise and fall times of the echo signals were each made to be about 1 ms. (i.e. together they were 10% of the total echo period); this figure is compatible with the optimum slope found for attenuation signalling (see later Section 3.4).

Initial tests made with permanent (i.e. non-switched) single echoes and anti-phase doublets* with delay times between 1 and 4 ms demonstrated the large tolerance of the ear to these affects; a relative fixed echo level of −10 dB was rated as acceptable by most observers. Average subjective results obtained from switching echoes at 25 Hz are shown in Fig. 3. It should be noted that the single-echo reverberation was not made zero-sum by simultaneously reducing the level of the main signal during its transmission; also, the results for zero-sum doublet transmission are for the case where the leading echo component was in anti-phase with the main signal but tests showed that inverting the sense of the doublet by putting the lagging echo in anti-phase with the main signal did not significantly modify the results. As expected, switched echoes were found to produce more programme impairment than the same level of permanent echo; the latter introduces a permanent comb-filter perturbation of the spectrum whereas switched echoes produce additional audible sidebands, pure tones being the most susceptible in this respect.

It was thought that an echo pattern which gave only phase perturbation of the programme spectrum would give minimal interferance even during switching. A circuit arrangement for switching such an echo pattern is shown in Fig. 4; this is a transversal filter with an all-pass characteristic. With this arrangement, however, it was found that the amplitude components of the switching interference was no less than in the above tests.

Some general conclusions can be drawn from the results shown in Fig. 3.

(1) There is no subjective advantage in using doublet echoes for signalling especially as they are no easier to detect than single echoes.

(2) Piano music is more susceptible to interference from echoes than other programme material.

(3) For a given echo level, the subjective rating of programme impairment increases with echo delay over the range 1 to 4 ms.

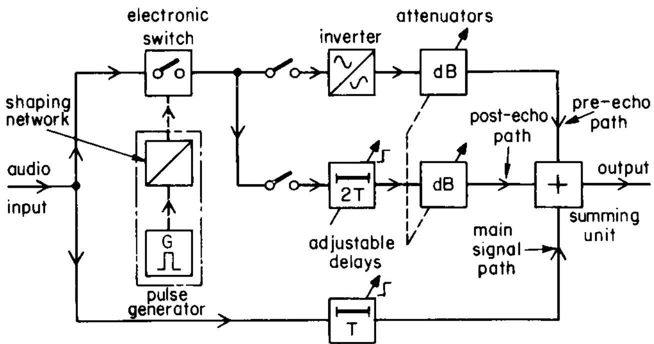

Fig. 2 - Experimental arrangement used for investigating reverberation signalling

* Telegraph signals can be simulated by using pseudo-random sequences; in the work described in this report, regular signals were found to represent the worst case and were therefore used in preference to pseudo-random sequences.

* It was thought that a zero-sum echo doublet (anti-phase pair) could be advantageous because it affects primarily only the phase component of the sound spectrum; modulation of the amplitude component is a second-order effect.

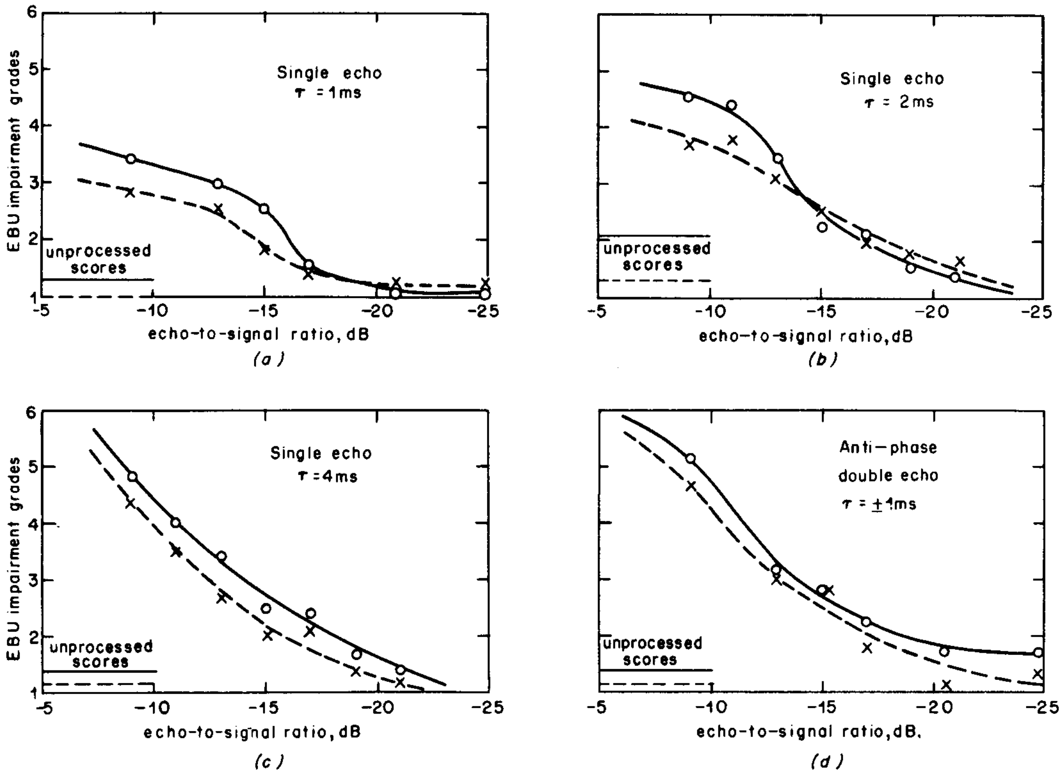

Fig. 3 - Subjective assessment of programme impairments resulting from the introduction of signal echoes at a switching rate of 25 Hz with unity mark-to-space ratio

(*a*) Single post-echo, T = 1 msec (*b*) Single post-echo, T = 2 msec (*c*) Single post-echo, T = 4 msec (*d*) Antiphase echo-doublet, T = 1 msec

—o——o— piano music --x---x-- male speech

(4) With a single echo seperated by 1–4 msecs and switched at 25 Hz, the 'just perceptible' level is between −15 dB and −20 dB relative to the main signal for most programmes; however it should be noted that with piano music, and an echo level of −15 dB, one third of the observers scored at least Grade 3.

These results show that, from the standpoint of programme impairment, reverberation switching offers a very energetic method of signalling; a practical method of detection has yet to be evolved however and wanted echoes would have to be separable from actual programme echoes.

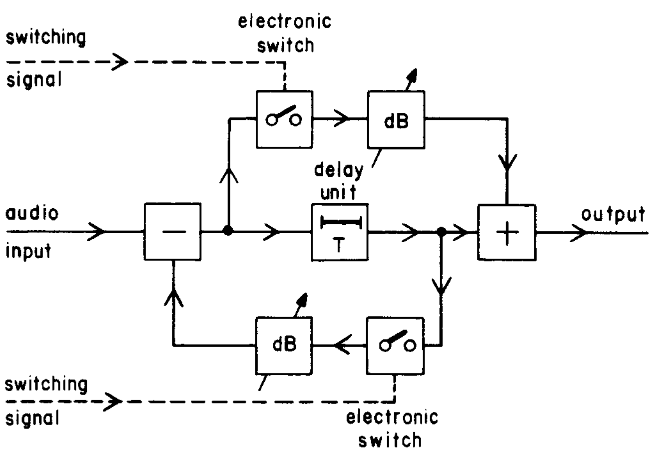

Fig. 4 - Switched transversal all-pass network for 'constant amplitude' reverberation signalling

3.4. Attenuation signalling

The object of this part of the investigation was to determine the subjective impairment of programme signals when short periods of high attenuation ('drop-outs') are regularly introduced. Subjective tests were carried out in order to determine the optimum parameters for maximum signalling rate consistent with acceptable programme impairment. The voltage controlled attenuator employed a shunt f.e.t. (Fig. 5) driven from a pulse generator whose output pulse could be varied in height, duration, repetition frequency and rise- and fall-times. This circuit gave over 30 dB of attenuation for a voltage drive of 2 volts d.a.p. with an approximately linear control of attenuation over a range of 20 dB.

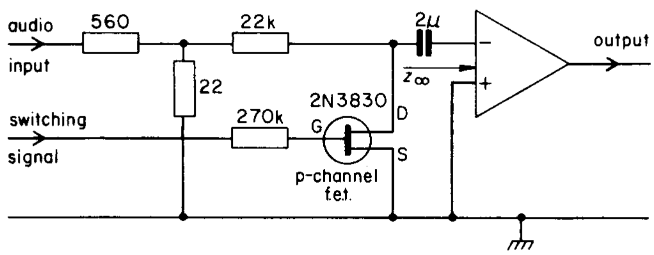

Fig. 5 - Simplified voltage-controlled attenuator used for investigating attenuation signalling

Initial tests with switched drop-outs showed that pure tones and signals in which discrete frequencies predominated were most sensitive to impairment. The majority of

the subjective tests were therefore confined to piano music containing slow passages of single notes.

The tests were arranged to explore the subjective effects of drop-out duration, rate, depth and rise/fall characteristics. Drop-outs restricted to treble components only (high-band) and bass components only (low-band) were also investigated.

The important results are shown in Figs. 6, 7, 8 and 9 which show average impairment scores. Some additional tests were also made in order to investigate the possibility of adding tone-bursts to compensate for (i.e. fill in) the drop-outs; although the shape of these signals were exactly complementary to the shape of the drop-outs, they had the effect of increasing rather than decreasing the impairment.

The results can be summarised as follows:

(1) The optimum drop-out duration (−3 dB) for both occasional and regular interruptions is between 1 and 2 ms; drop-outs shorter than this produce audible 'clicks' whereas longer ones give rise to objectionable 'thumps' (Fig. 6).

(2) The maximum drop-out rate for imperceptible impairment is between 1 and 2 Hz for depths greater than −20 dB and durations around 1·5 ms (Fig. 7).

(3) Regular drop-outs are perceptible only when their depth exceeds about −4 dB, but for depths exceeding about −12 dB the annoyance does not appear to increase significantly.

(4) The shape of the drop-out was found to be less important than the maximum rate of descent or rise; the maximum rate for 'just perceptible' impairment was about 12 μs per dB (Fig. 8). No advantage was obtained by using raised-cosine drop-outs (Fig. 9); nor was any advantage to be found in the use of asymmetrical rise/fall characteristics.

(5) There is a small advantage to be gained by restricting drop-outs to only part of the signal spectrum; for example, a high-band system can offer about 1 grade of improvement and a low-band system about ½ grade using a crossover frequency of 1 kHz (Fig. 7).

It may be concluded that attenuation signalling can be made subliminal but the information rate has to be limited to the low value of 1 or 2 bits/sec. Furthermore, there are severe practical difficulties associated with drop-out detection. It must be concluded that this form of subliminal signalling would have a low probability of success.

3.5. Frequency-notch signalling

The object here was to explore the possibilities of subliminal signalling by processing the spectrum of the programme signal.

Signalling by switching frequency notches (or peaks) in and out of the programme spectrum was ruled out on account of the limited potential information rate. (Signalling must await the presence of a particular frequency in the programme signal.) Also, previous reported work[8] had shown that 'frequency peaks' generally cause more programme interference than equivalent 'frequency notches'.

Work was therefore concentrated on determining the subjective effects of *permanent* frequency notches in the programme signal; the programme-free locations so obtained, might allow additive low-level signalling within the

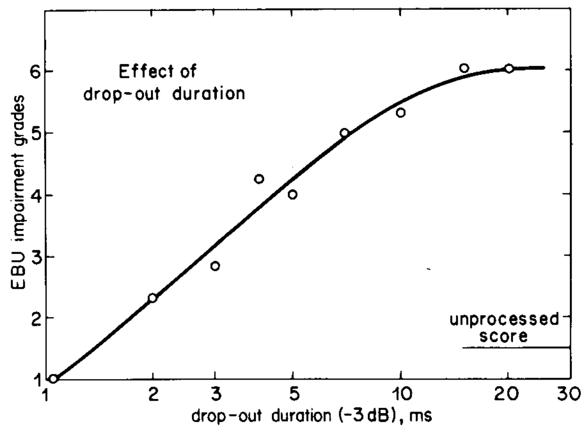

Fig. 6 - Attenuation signalling : mean variation of subjective programme interference with dropout duration (msecs). Drop-out frequency, 2 Hz; depth, −25 dB; rise/fall time, 300 µsecs

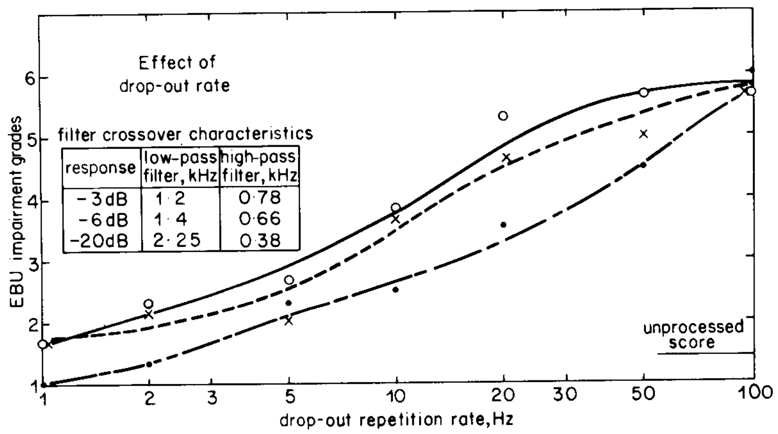

Fig. 7 - Attenuation signalling : mean variation of subjective programme interference with frequency of drop-outs showing comparison with low-band and high-band cases (crossover 1 kHz). Drop-out depth, −25 dB; drop-out duration (−3 dB), 1·5 msec; drop-out rise/fall time, 300 µsecs linear

full bandwidth signal
low-band signal (f<1 kHz)
high-band signal (f>1 kHz)

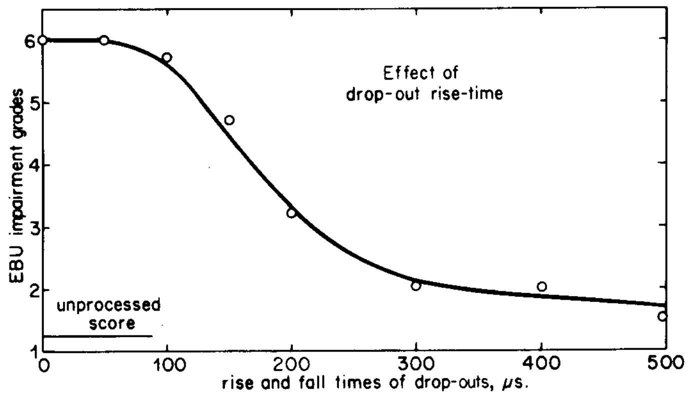

Fig. 8 - Attenuation signalling : mean variation of subjective programme interference with drop-out rise/fall time (linear, μsecs). Drop-out frequency, 2 Hz; depth, −25 dB; duration, 1·5 msecs.

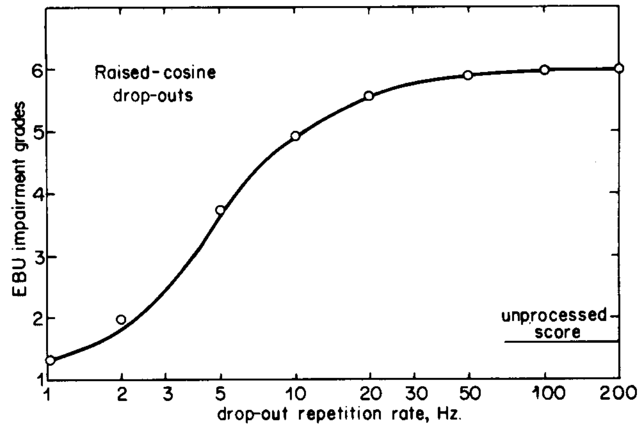

Fig. 9 - Attenuation signalling : mean variation of subjective programme interference with frequency for raised-cosine shaped drop-outs. Drop-out depth, −25 dB; duration 1·5 msecs

programme bandwidth.* The subjective tests were carried out in order to determine the optimum notch parameters consistent with maximum signalling rates and acceptable programme impairments.

Preliminary tests showed that it was impractical to use the frequency region below about 1 kHz and also confirmed that the spectral range which includes the fundamental frequencies of musical instruments (0 – 4·5 kHz) is the most susceptible to impairments caused by spectral notches; piano music, particularly in the region around 1 kHz, was found to be generally the most sensitive in this respect. The work was arranged to explore the subjective effects of frequency-notch depth, width, centre-frequency and also the effect of multiple notches.

The preliminary tests also indicated that the method might form the basis of a successful signalling system, and the possibility of applying the method to h.f. broadcasting appeared very promising. It was therefore decided to concentrate on the limited band 0 to 6 kHz. With a signal-to-noise ratio of 40 dB, so as to simulate typical h.f. broadcasting conditions, this had the benefit of restricting the number of subjective tests to a manageable figure and had the additional advantage of a particular practical application. Moreover, it was thought that, if a signalling system could be developed for use with the lower audio frequencies, it should be relatively easy to apply the same principles to wider band systems; the results described in this report tend to support this belief.

The main results of the subjective tests are shown in Figs. 10 and 12 which give the impairment scores averaged over all observers.

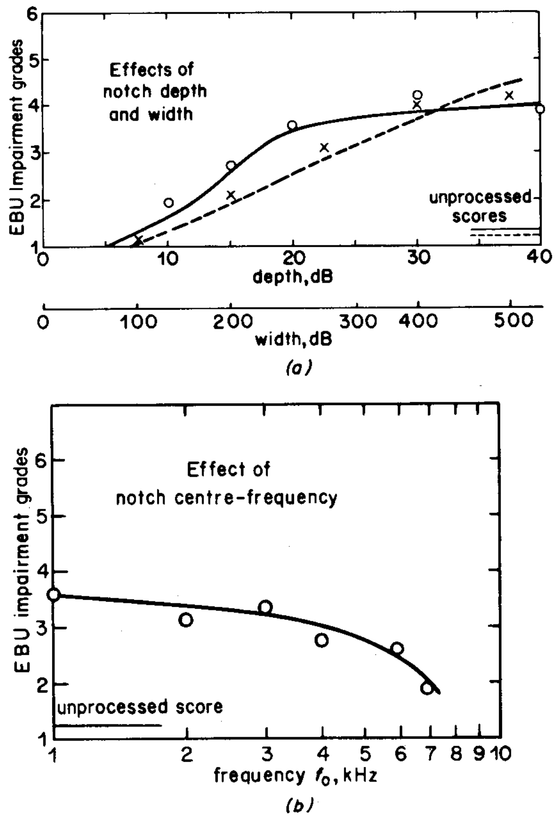

Fig. 10 - Frequency-notch signalling showing characteristics for centre frequencies offset by quarter-tones

(*a*) mean variation of subjective programme interference with notch depth and width (−3 dB) at 1 kHz

—o——o— variation with depth (Δf ≃ 200 Hz)

--x---x-- variation with width (depth ≃ −50 dB)

(*b*) mean variation of subjective programme interference with centre-frequency for a single notch of fixed depth and bandwidth ±5 Hz at −40 dB

* An example of an *out-of-band* additive system is provided by the Piccolo method of multitone signalling.[26]

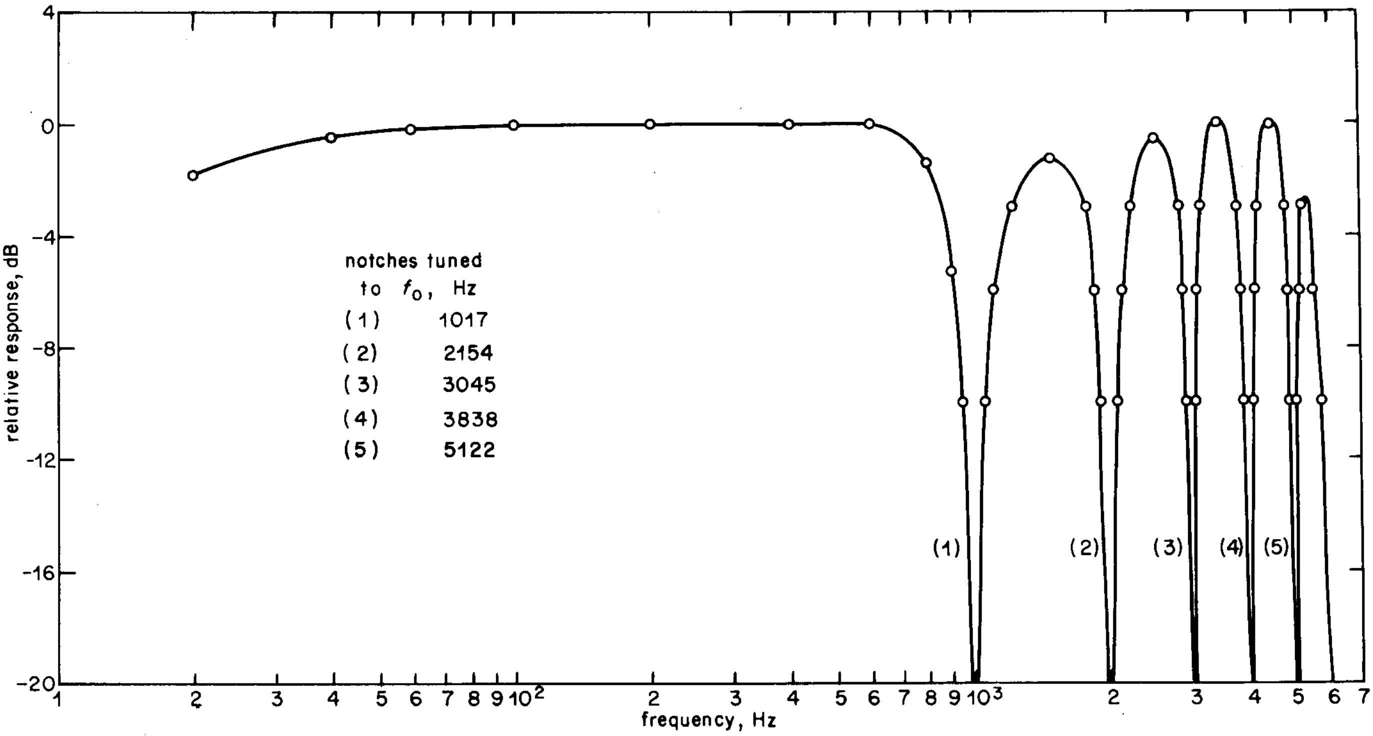

Fig. 11 - Frequency-notch signalling : frequency characteristic used for investigating the programme impairments produced by the presence of multiple notches placed in a 6 kHz sound channel at optimum centre-frequencies

The more important results of this investigation may be summarised as follows:

(1) Tests with piano music showed that deep frequency notches tuned to fundamental frequencies on the musical scale can produce quite unacceptable programme impairment.

(2) Single notches which are carefully offset by a quarter tone from the musical scale can be made almost totally imperceptible for most programme material; electronic music is a notable exception.

The following results 3 to 7 are for notches offset by a quarter-tone.

(3) For piano music the depth of a single frequency notch has to be greater than about 10 to 15 dB for more than 50% of observers to rate the programme impairments as just perceptible (Grade 2); also, the impairments do not increase much beyond Grade 3½ for notch depths greater than −20 dB (Fig. 10(*a*)).

(4) The impairments increase almost linearly as the frequency notch is widened. Fig. 10(*a*) shows that an increase of 100 Hz in the 3 dB-width of a 50 dB-deep notch produces almost one grade change in impairment.

(5) Over the frequency range 1 to 6 kHz the effect of varying the notch centre-frequency is quite small; the impairment falls off with frequency and is most severe around 1 kHz (Fig. 10(*b*)). At frequencies above 7 kHz the impairments tend to imperceptibility but this range has not been explored extensively.

(6) It was found practicable to include simultaneously up to 5 notches within the frequency band 1 to 6 kHz but for acceptable impairments their centre-frequencies must be carefully chosen to be offset from the fundamental and harmonic tones of the musical scale.

(7) Fig. 12 (with Fig. 11) gives the subjective results in the form of histograms for a 5-notch system with 'offset' centre-frequencies suitable for a 6 kHz sound channel. A comparative scale (Table 3) of impairment assessment was used as it was thought to be more sensitive than absolute ratings in this case. In the worst case (piano music (b)) only about 30% of listeners thought that the frequency-notched programme was significantly worse (i.e. scored ⩽−2) than the unprocessed programme; for male speech this percentage was slightly less and, for dance music, the surprising result is that about 10% of listeners actually preferred the processed material!

Subliminal signalling based on the presence of deep spectral notches inserted into the programme signal thus appears to be feasible. We have shown that isolated notches as deep as −50 dB at the centre-frequency and with a bandwidth of ±5 Hz at −40 dB will not produce undue programme impairment (Grade 3) if they are positioned in the frequency range 1 to 6 kHz provided that the centre-frequencies are carefully offset from the musical scale; for frequencies beyond 6 kHz it appears that spectral notching will be totally acceptable but more comprehensive tests are required to confirm this.

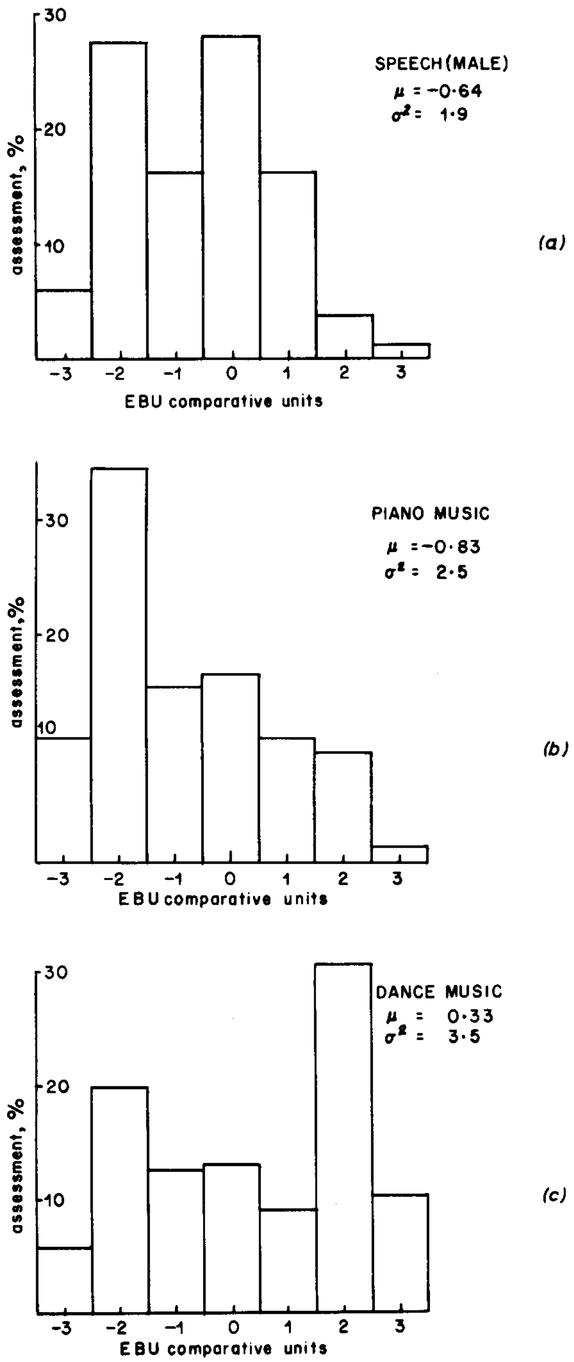

Fig. 12 - Frequency-notch signalling : histogram analysis of subjective programme interference produced by processing a 6 kHz sound signal with the transfer characteristic given by Fig. 11 (unweighted peak signal-to-r.m.s. noise ratio of 40 dB)

(*a*) speech (male) (*b*) piano music (*c*) dance music

μ = mean score σ^2 = variance of scores

Multiple notches of the same order of magnitude can be placed as close as 1 kHz apart. For low performance sound circuits of restricted bandwidth (say 6 kHz with 40 dB signal-to-noise ratio) the resulting impairments to programme are just acceptable if 5 notches are employed; only 30% of listeners are likely to notice any disturbance to programme with the most exacting programme material. It is therefore not too sanguine to expect signalling rates up to 40 or 50 bauds with multitone frequency-notch systems.

4. Final remarks and conclusions

The feasibility of simultaneous subliminal signalling in sound circuits has been explored both theoretically and experimentally. The theoretical approach was to examine the various well-known, and also lesser-known, limitations in human hearing and to estimate whether their exploitation could lead to a practical subliminal signalling system.

From a list of sixteen theoretical systems, which is not claimed to be exhaustive, four systems were thought to have sufficient probability of success to justify experimental investigation; they all involve modulation of the actual programme signal. After subjective tests only three of these systems were seen to offer useful subliminal signalling speeds; the main subjective characteristics of these systems are summarised below:

1. Reverberation signalling : employs echoes or anti-phase echo doublets which are switched into the sound signal by the message in digital form; echoes of 1 ms duration at a level of −15 dB relative to the main signal can signal at 50 bauds with only 30% of listeners perceiving impairment; a satisfactory process of signal detection has yet to be developed.

2. Attenuation signalling : the sound signal is attenuated to a very low level by pulses representing the message; for acceptable results, the information rate must not exceed 2 bauds with specially shaped drop-outs of 1 to 2 ms duration; although a peak attenuation greater than 20 dB can be tolerated, the signalling speed is low and a satisfactory detection process is not yet developed.

3. Frequency-notch signalling : the sound signal is permanently processed by inserting one or more notches into its spectrum in order to allow simultaneous low-level modulated tones to be inserted; isolated notches with a ±5 Hz bandwidth at −40 dB do not produce noticeable impairments if their centre-frequencies are carefully chosen. Multiple notches can be employed with a minimum spacing of 1 kHz; in a low-quality 6 kHz circuit only 30% of listeners are likely to notice any disturbing effects. Signalling rates up to 40 bauds appear possible.

The main conclusion is that signalling methods based on frequency-notch processing are likely to be more practical and offer much greater potential subliminal information rates (perhaps 40 bauds) than the method of attenuation drop-outs (2 bauds). Reverberation signalling on the other hand can provide an extremely powerful subliminal signalling system (50 bauds) but a satisfactory detection process has yet to be evolved. The experimental work on frequency-notch signalling was concentrated on examining subjective performance in the lower but more sensitive end of the audio band (0 to 6 kHz); more subjective tests are required to confirm the potentiality of using the upper half-spectrum of a high-quality sound circuit (0 to 15 kHz).

Finally, it is recommended that, based on the present work, the practical aspects of multitone frequency-notch signalling should be studied in greater depth and also that the feasibility of detecting reverberation signalling should be carefully considered.

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